Application Notes for Configuring Ascom i62 VoWiFi Handsets with Avaya Aura® Communication Manager and Avaya Aura® Session Manager – Issue 1.0

Abstract

These Application Notes describe the configuration steps for provisioning Ascom's i62 VoWiFi handsets to interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as any observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the configuration steps for provisioning Ascom's i62 VoWiFi (i62) handsets to interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. Ascom's i62 handsets are configured to register with Session Manager and are configured with 9640 SIP endpoint template. The Ascom i62 handsets then behave as third-party SIP extensions on Communication Manager. The handsets are able to make/receive internal and PSTN/external calls, and have full voicemail and other telephony features available on Communication Manager.

2. General Test Approach and Test Results

The interoperability compliance testing evaluates the ability of Ascom i62 VoWiFI handsets to make and receive calls to and from Avaya H.323, Avaya SIP, and PSTN endpoints. Avaya Aura® Messaging was used to allow users leave voicemail messages and to demonstrate Message Waiting Indication and DTMF on the Ascom i62 handsets.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in these DevConnect Application Notes included the enablement of supported encryption capabilities in the Avaya products. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

Support for these security and encryption capabilities in any non-Avaya solution component is the responsibility of each individual vendor. Readers should consult the appropriate vendor-supplied product documentation for more information regarding those products.

For the testing associated with these Application Notes, the interface between Avaya systems and Ascom i62 VoWiFi handsets did not include use of any specific encryption features as requested by Ascom.

Avaya's formal testing and Declaration of Conformity is provided only on the headsets/Smartphones that carry the Avaya brand or logo. Avaya may conduct testing of non-Avaya headset/handset to determine interoperability with Avaya phones. However, Avaya does not conduct the testing of non-Avaya headsets/Smartphones for: Acoustic Pressure, Safety, Hearing Aid Compliance, EMC regulations, or any other tests to ensure conformity with safety, audio quality, long-term reliability or any regulation requirements. As a result, Avaya makes no representations whether a particular non-Avaya headset will work with Avaya's telephones or with a different generation of the same Avaya telephone.

Since there is no industry standard for handset interfaces, different manufacturers utilize different handset/headset interfaces with their telephones. Therefore, any claim made by a headset vendor that its product is compatible with Avaya telephones does not equate to a guarantee that the headset will provide adequate safety protection or audio quality

2.1. Interoperability Compliance Testing

The compliance testing included the test scenarios shown below. Note that when applicable, all tests were performed with Avaya SIP, Avaya H.323, Ascom i62 and PSTN endpoints.

- Basic Calls
- Media Shuffling
- Session Refresh Timer
- Long Duration Call
- Hold and Long Hold, Retrieve and Brokering (Toggle)
- Feature Access Code dialing with special characters
- Attended and Blind Transfer
- Call Forwarding Unconditional, No Reply and Busy
- Call Waiting
- Call Park/Pickup
- EC500, where Avaya deskphone is the primary phone and i62 handset being the EC500 destination.
- Multi-Device Access (MDA)
- Attended Conference (also local three-way calling)
- Do Not Disturb
- Calling Line Name/Identification
- Codec Support (G.711, G.729, G.722)
- DTMF Support
- Coverage
- Voice Mail, Message Waiting Indication
- Serviceability

2.2. Test Results

The tests were all functional in nature and performance testing was not included. All test cases passed successfully with the following observations/limitations noted below:

1. When using the EC500 (concurrent call) feature, if an i62 handset or an Avaya endpoint answers the call before two rings, the call is dropped. This is due to the "Cellular Voice Mail Detection" field default value seen in "off-pbx-telephone configuration-set" form of Communication Manager. The default value for this field is "timed (seconds): 4" which means that if Communication Manager receives an answer within 4 seconds then it will be considered as the cellular voicemail picking up the call, and so call will be dropped and proceed to do Communication Manager coverage processing instead. The workaround is to answer the call after 2 rings, or change the "Cellular Voice Mail Detection" field value to "none" or decrease "timed" value. Note that changing the "off-pbx-telephone configuration-set" affects all users in the same set, so if cellular users are grouped with i62 handset users, calls may be answered by a cellular user's voicemail instead of following the coverage criteria in Communication Manager.
2. When an i62 handset is configured as an EC500 destination for an Avaya endpoint, an incoming call to the Avaya endpoint will ring both the Avaya endpoint and the i62 handset. When the call is declined on the i62 handset, the Avaya endpoint continues to ring as per normal design.
3. All compliance testing was done using UDP and TCP (preferred) as the transport protocol.
4. Negotiation of G.722 between endpoints, such as the Ascom i62, requires support for the codec to be configured on Communication Manager.
5. Diversion can be accomplished via Feature Access Code configured in Communication Manager or locally via i62 handset setting. The call forward on busy feature however, can only be accomplished via Communication Manager with 486 BUSY HERE automatically sent to the caller by Communication Manager on behalf of the busy endpoint per design.
6. When an Avaya endpoint or an i62 handset calls another i62 handset, after the called i62 handset declines the call, the display for the i62 calling party shows busy whereas the Avaya calling party receives the busy tone.
7. In the blind transfer scenario involving three i62 handsets, where A is the calling, B is the called and transfer-from and C is the transfer-to parties. The display on the transfer-to party C showed "Redirected x y" where "x" is the extension number of the calling party A and "y" is the name of the transfer-from party B. According to Ascom, transfer-to party C displayed information conveyed by Communication Manager, and "redirected" was displayed because the INVITE included a History-Info header (Ascom ticket: MRS-145).
8. Ascom i62 handset supports third party conference, which is, i62 makes two calls simultaneously and conferences the calls locally.
9. When multiple voice messages are left for an i62 handset, the handset shows the total number of messages as only "1" in the display even though there are multiple messages. This is because there is no counter information sent in the NOTIFY from Avaya Aura® Messaging.

10. For Multi-Device Access (MDA) feature in i62 handset to function, i62 needs to be configured using and registering through Endpoint ID. Also the MWI configuration has to be identical on all i62 handsets that are configured for MDA. Refer to **Section 7.3** for details.
11. Per design, i62 handsets do not have a redial button. User needs to use “Call List” and redial the numbers.
12. When outgoing calls are configured to be restricted for an i62 handset on Communication Manager, the i62 display showed “No Channel Available” when user attempted to make an outbound call.
13. When the Ethernet to uplink Wireless Access Point was disconnected and connected back after 60 secs, it took more than 5 mins for the i62 handsets to re-register and able to make new calls. Expected behavior had been that the Ascom i62 would to try to re-register as soon as an outgoing call fails due to no response from Session Manager. In this case, the workaround was to restart the phones and get immediate connectivity. Active calls will stay up until session refresh expires when the link to the Session Manager is down.

2.3. Support

Technical support for the Ascom i62 wireless handsets can be obtained through a local Ascom supplier or Ascom global technical support:

- Email: support@ascom.com
- Help desk: +46 31 559450

3. Reference Configuration

Figure 1 shows the network topology during compliance testing. The Ascom i62 VoWiFi handsets connect to an Ascom approved wireless router which is placed on the LAN. The i62 handsets register with Session Manager in order to be able to make/receive calls with the Avaya H.323 and SIP endpoints on Communication Manager and with the PSTN. The handsets are configured by Ascom Windows Portable Device Manager (WinPDM) using the Ascom DeskTop Programmer DP1.

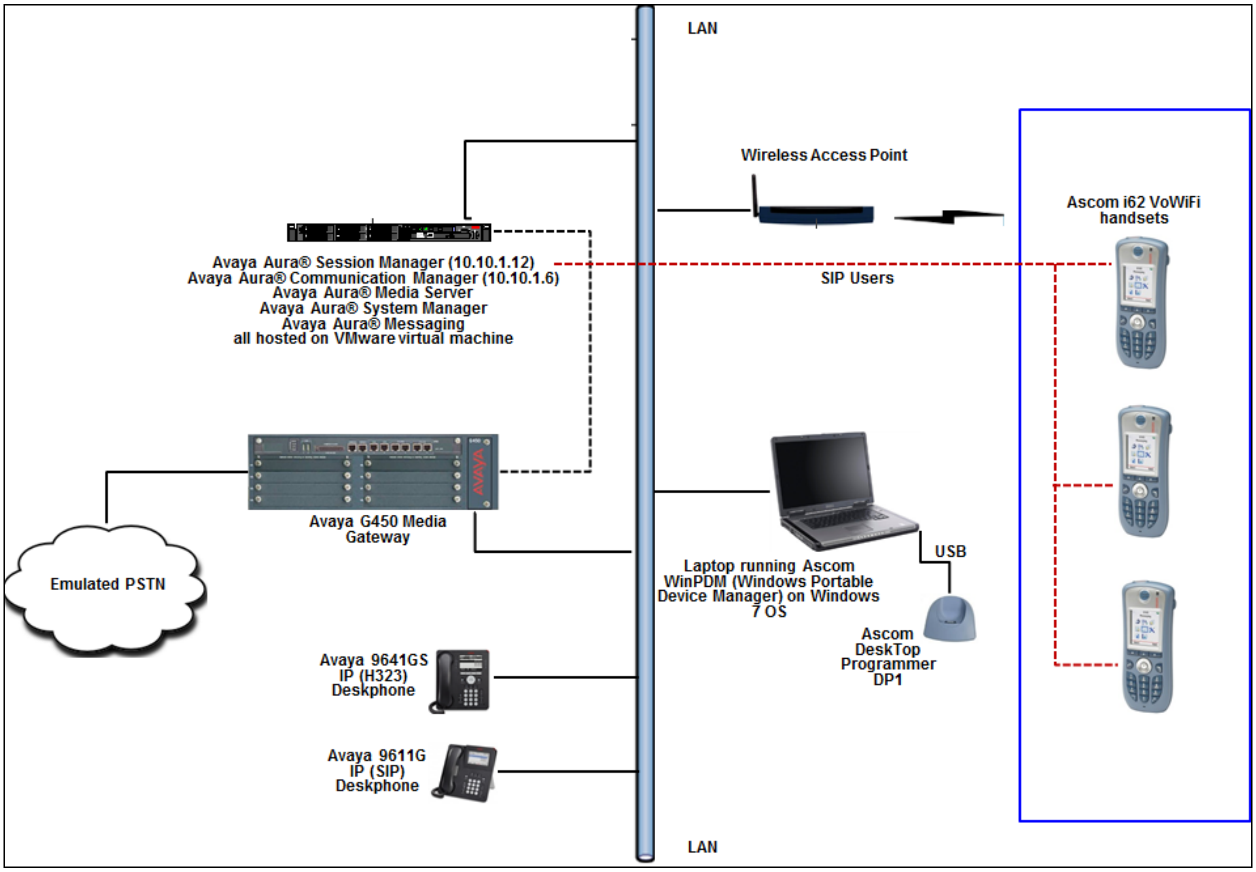


Figure 1: Network Solution of Ascom i62 VoWiFi Handsets with Avaya Aura® Communication Manager and Avaya Aura® Session Manager

4. Equipment and Software Validated

The following equipment and software was used for the compliance test.

Equipment/Software	Release/Version
Avaya Aura® Communication Manager running on Virtual Server	7.1.2.0.0-FP2
Avaya Aura® Session Manager running on Virtual Server	7.1.2.0.712004
Avaya Aura® System Manager running on Virtual Server	7.1.2.0 (Feature Pack 2)
Avaya Aura® Messaging running on Virtual Server	07.0.0.0.441
Avaya G450 Gateway	38 .18 .0 /1
Avaya IP Deskphones: • 9641GS (H.323) • 9611G (SIP)	 6.6506 7.1.1.0.9
Ascom Windows Portable Device Manager running on Windows 7	3.11.1
Ascom DeskTop Programmer D1	N/A
Ascom i62 VoWiFi handsets	6.0.6

5. Configure Avaya Aura® Communication Manager

It is assumed that a fully functioning Communication Manager is in place with necessary licensing and connecting to Session Manager via SIP Trunk. For further information on the configuration of Communication Manager please see **Section 10** of these Application Notes. The following sections go through the following.

- Dial Plan Analysis
- Feature Access Codes
- Network Region
- IP Codec
- Coverage Path/Hunt Group

Ensure that the SIP endpoints license is valid as shown below by using the command **display system-parameters customer-options**.

```
display system-parameters customer-options                          Page   1 of  12
                                OPTIONAL FEATURES

     G3 Version: V17                                  Software Package: Enterprise
       Location: 2                                     System ID (SID): 1
       Platform: 28                                    Module ID (MID): 1

                                                                  USED
                                Platform Maximum Ports: 48000 168
                                      Maximum Stations: 36000 44
                              Maximum XMOBILE Stations: 36000 0
                    Maximum Off-PBX Telephones - EC500: 41000 2
                    Maximum Off-PBX Telephones -   OPS: 41000 20
                    Maximum Off-PBX Telephones - PBFMC: 41000 0
                    Maximum Off-PBX Telephones - PVFMC: 41000 0
                    Maximum Off-PBX Telephones - SCCAN: 0     0
                         Maximum Survivable Processors: 313   1
```

Note: A printout of the Signalling and Trunk groups that were used during compliance testing can be found in the **Appendix** of these Application Notes.

5.1. Configure Dial Plan Analysis

Use the **change dialplan analysis** command to configure the dial plan using the parameters shown below. Extension numbers (**ext**) are those beginning with **33** and **34**. Feature Access Codes (**fac**) use digits **8** and **9**. Dial Access Codes (**dac**) use characters ***** or **#**.

```
change dialplan analysis                                     Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location: all           Percent Full: 5

    Dialed    Total  Call      Dialed    Total  Call      Dialed    Total  Call
    String    Length Type      String    Length Type      String    Length Type
   33           4    ext
   30           4    aar
   33           4    ext
   8            1    fac
   9            1    fac
   *            3    dac
   #            3    dac
```

5.2. Configure Feature Access Codes

Use the **change feature-access-codes** command to configure access codes which can be dailed from i62 handsets to initiate Communication Manager call features. These access codes must be compatible with the dial plan described in **Section 5.1**. Some of the access codes configured during compliance testing are shown below.

```
change feature-access-codes                                     Page   1 of  10
                               FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code:
         Abbreviated Dialing List2 Access Code:
         Abbreviated Dialing List3 Access Code:
Abbreviated Dial - Prgm Group List Access Code:
                      Announcement Access Code: *05
                       Answer Back Access Code: 007
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code: 8
    Auto Route Selection (ARS) - Access Code 1: 9     Access Code 2:
                 Automatic Callback Activation:        Deactivation:
Call Forwarding Activation Busy/DA: *07    All: *06    Deactivation: *16
   Call Forwarding Enhanced Status:        Act:        Deactivation:
                         Call Park Access Code: 008
                       Call Pickup Access Code: *09
CAS Remote Hold/Answer Hold-Unhold Access Code: *10
                  CDR Account Code Access Code: *11
                        Change COR Access Code:
                   Change Coverage Access Code:
            Conditional Call Extend Activation:        Deactivation:
                   Contact Closure   Open Code:          Close Code:
```

5.3. Configure Network Region

Use the **change ip-network-region x** (where x is the network region to be configured) command to assign an appropriate domain name to be used by Communication Manager, in the example below **bvwdev.com** is used. Note that this domain is also configured in **Section 6.1**of these Application Notes.

```
change ip-network-region 1                                     Page   1 of  20
                               IP NETWORK REGION
  Region: 1       NR Group: 1
Location: 1       Authoritative Domain: bvwdev.com
    Name: Loc-1                     Stub Network Region: n
MEDIA PARAMETERS                    Intra-region IP-IP Direct Audio: yes
      Codec Set: 1                  Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                          IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                       RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

5.4. Configure IP Codec

Use the **change ip-codec-set x** (where x is the IP codec set used) command to designate a codec set compatible with the i62 handsets. During compliance testing the codecs **G.711MU**, **G.729** and **G.722-64K** were tested.

```
change ip-codec-set 1                                           Page   1 of   2

                          IP MEDIA PARAMETERS
    Codec Set: 1

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711MU           n         2        20
 2: G.729             n         2        20
 3: G.722-64K                   2        20
 4:
 5:
 6:
 7:

     Media Encryption                   Encrypted SRTCP: enforce-unenc-srtcp
 1: none
 2:
 3:
 4:
 5:
```

5.5. Configuration of Coverage Path and Hunt Group for Voicemail

The coverage path setup used for compliance testing is illustrated below. Note the following:

Don't' Answer is set to **y:**	The coverage path will be used in the event the phone set is not answered.
Number of Rings is set to **2:**	The coverage path will be used after 2 rings.
Point 1 is set to **h4**	Hunt Group 4 is utilised by this coverage path.

```
display coverage path 4
                                 COVERAGE PATH

                   Coverage Path Number: 4
     Cvg Enabled for VDN Route-To Party? n        Hunt after Coverage? n
                       Next Path Number:          Linkage

COVERAGE CRITERIA
    Station/Group Status    Inside Call     Outside Call
             Active?             n               n
               Busy?             y               y
       Don't Answer?             y               y         Number of Rings: 2
                All?             n               n
 DND/SAC/Goto Cover?             y               y
   Holiday Coverage?             n               n


COVERAGE POINTS
    Terminate to Coverage Pts. with Bridged Appearances? n
  Point1: h4              Rng:    Point2:
  Point3:                         Point4:
  Point5:                         Point6:
```

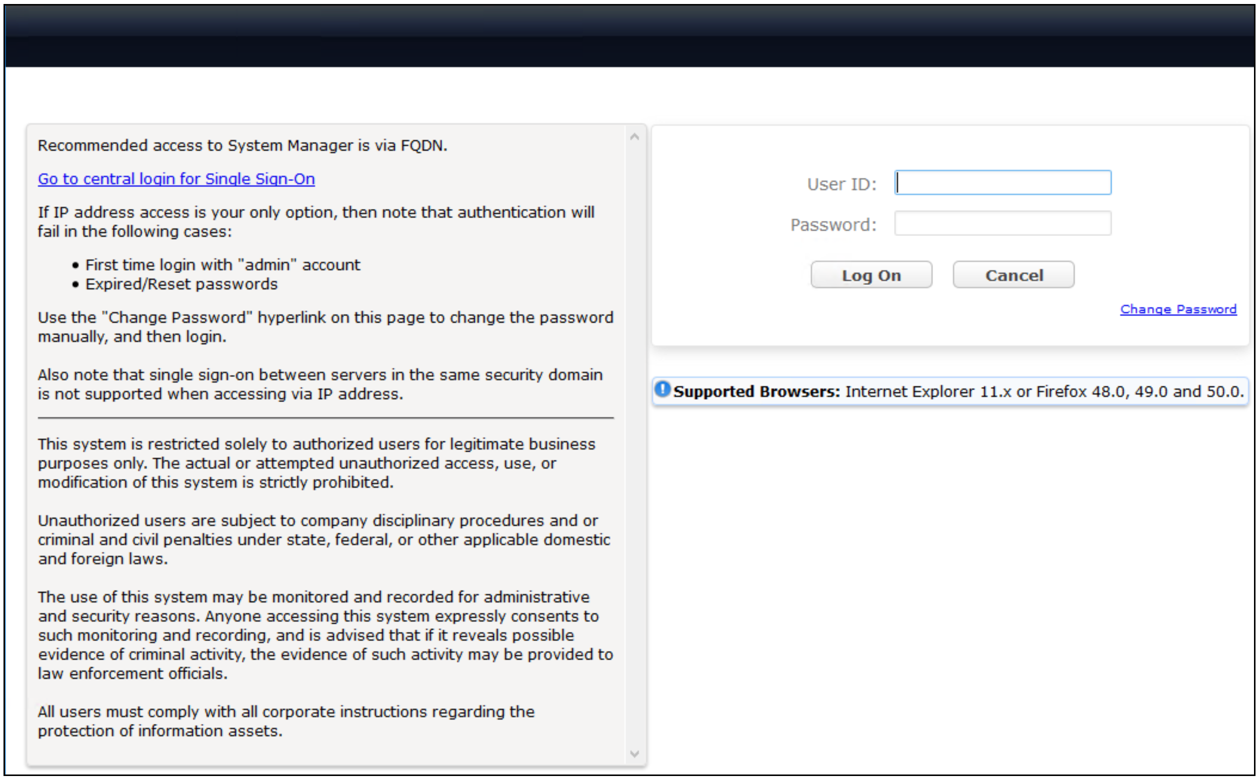
The hunt group used for compliance testing is shown below. Note that on **Page 1** the **Group Extension** is **3333,** which is used to dial for messaging and on **Page 2 Message Center** is set to **sip-adjunct**.

```
display hunt-group 4                                           Page   1 of   60
                                  HUNT GROUP

            Group Number: 4                              ACD? n
              Group Name: AMM                          Queue? n
         Group Extension: 3333                        Vector? n
              Group Type: ucd-mia               Coverage Path:
                      TN: 1         Night Service Destination:
                     COR: 1                  MM Early Answer? n
           Security Code:         Local Agent Preference? n
 ISDN/SIP Caller Display:
```

```
display hunt-group 4                                           Page   2 of   60
                                  HUNT GROUP




                      Message Center: sip-adjunct

     Voice Mail Number        Voice Mail Handle         Routing Digits
                                                 (e.g., AAR/ARS Access Code)
     3000                     3000
```

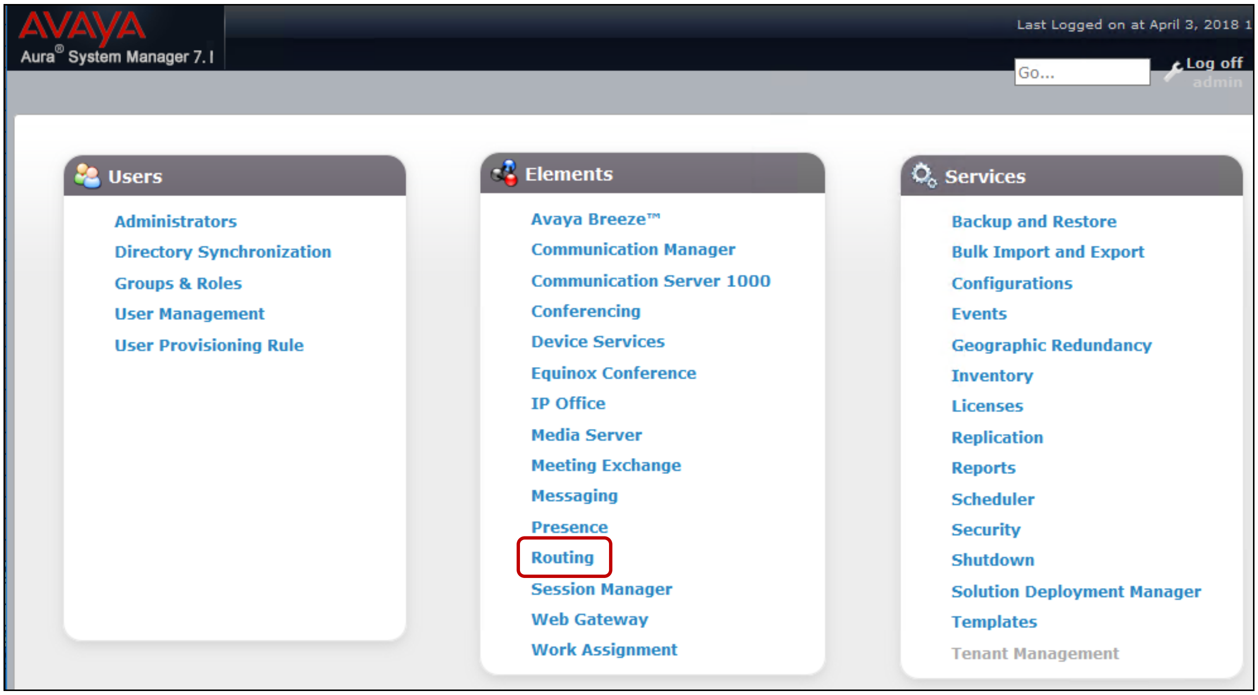
6. Configure Avaya Aura® Session Manager

The Ascom i62 VoWiFi handsets are added to Session Manager as SIP users. In order make changes in Session Manager a web session to System Manager is opened. Navigate to http://<System Manager IP address>/SMGR, enter the appropriate credentials and click on **Log On** as shown below.

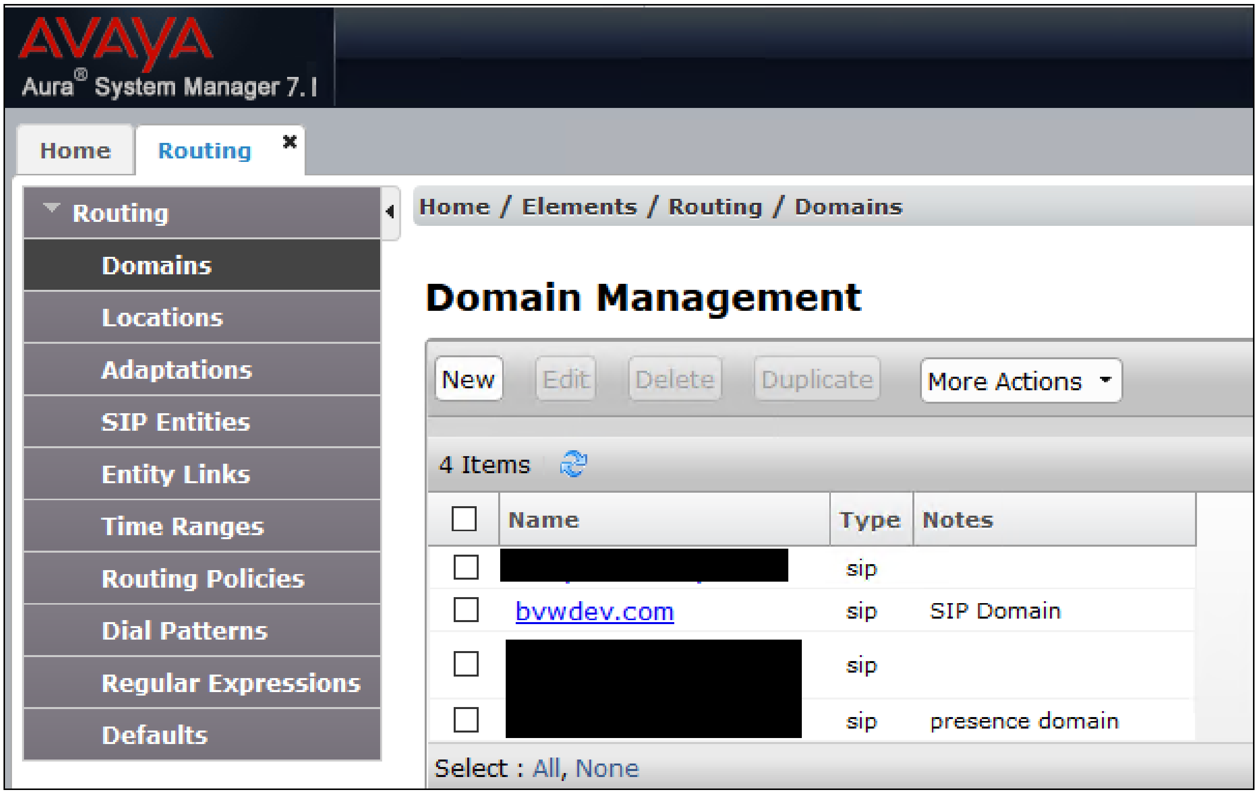


6.1. Configuration of a Domain

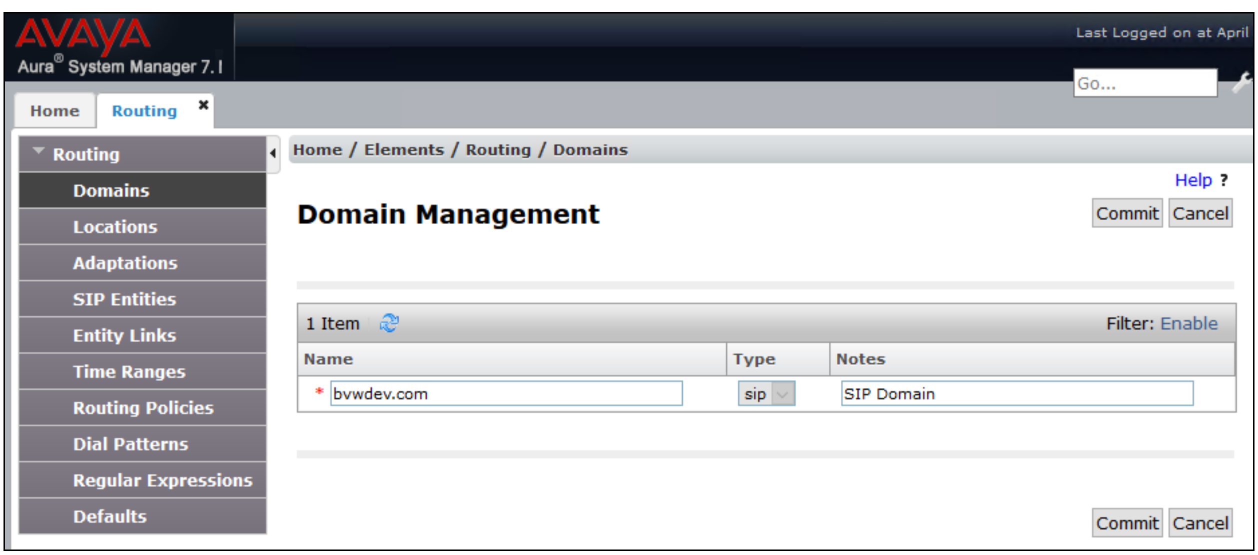
Click on **Routing** highlighted below.



Click on **Domains** in the left window. If there is not a domain already configured click on **New** to create a new domain name with **Type sip**. In the example below, there exists a domain called **bvwdev.com** which has been already configured.

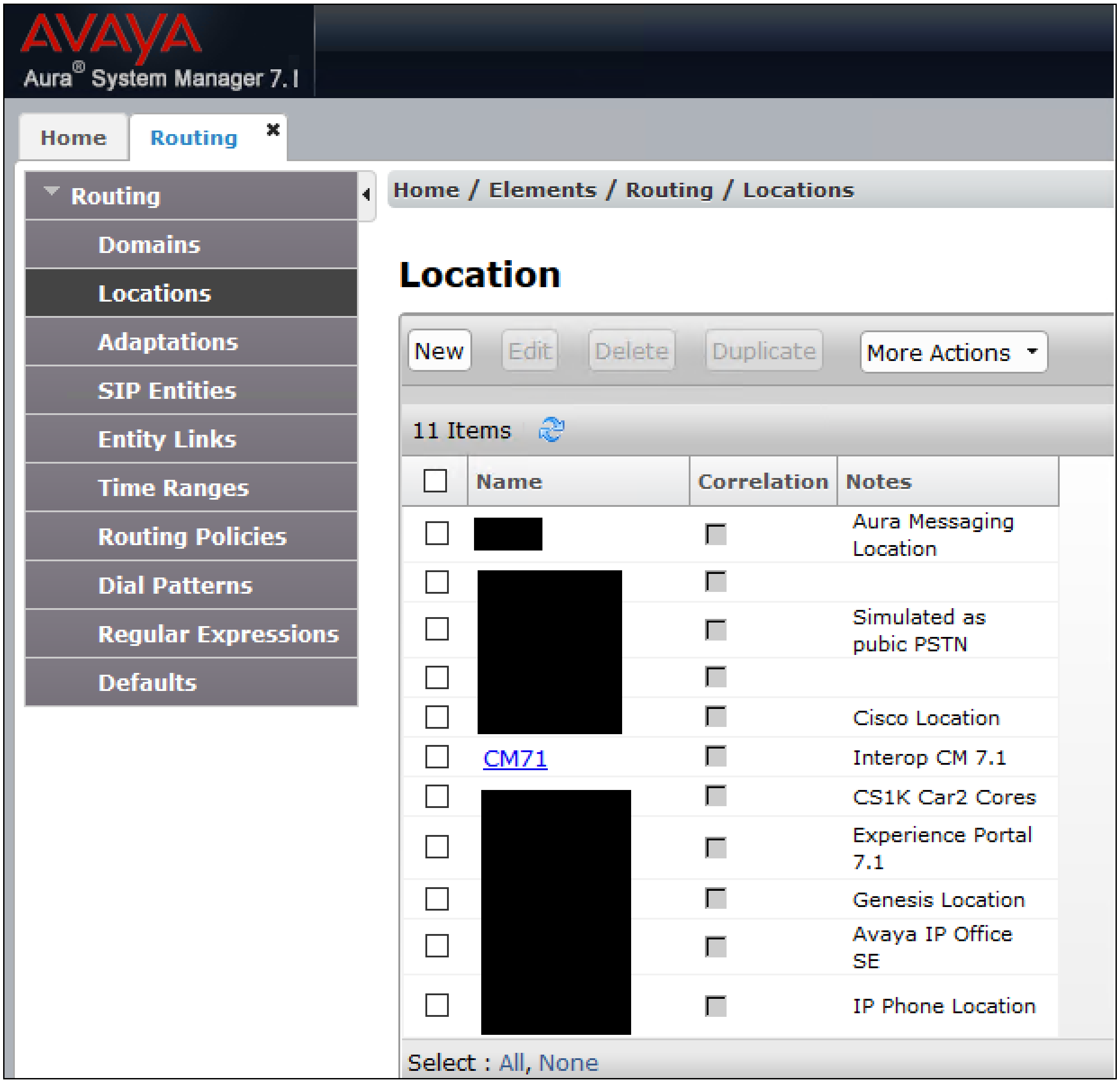


Clicking on the domain name above will open the following window; this is simply to show an example of such a domain. When entering a new domain the following should be entered, once the domain name is entered click on **Commit** to save this.

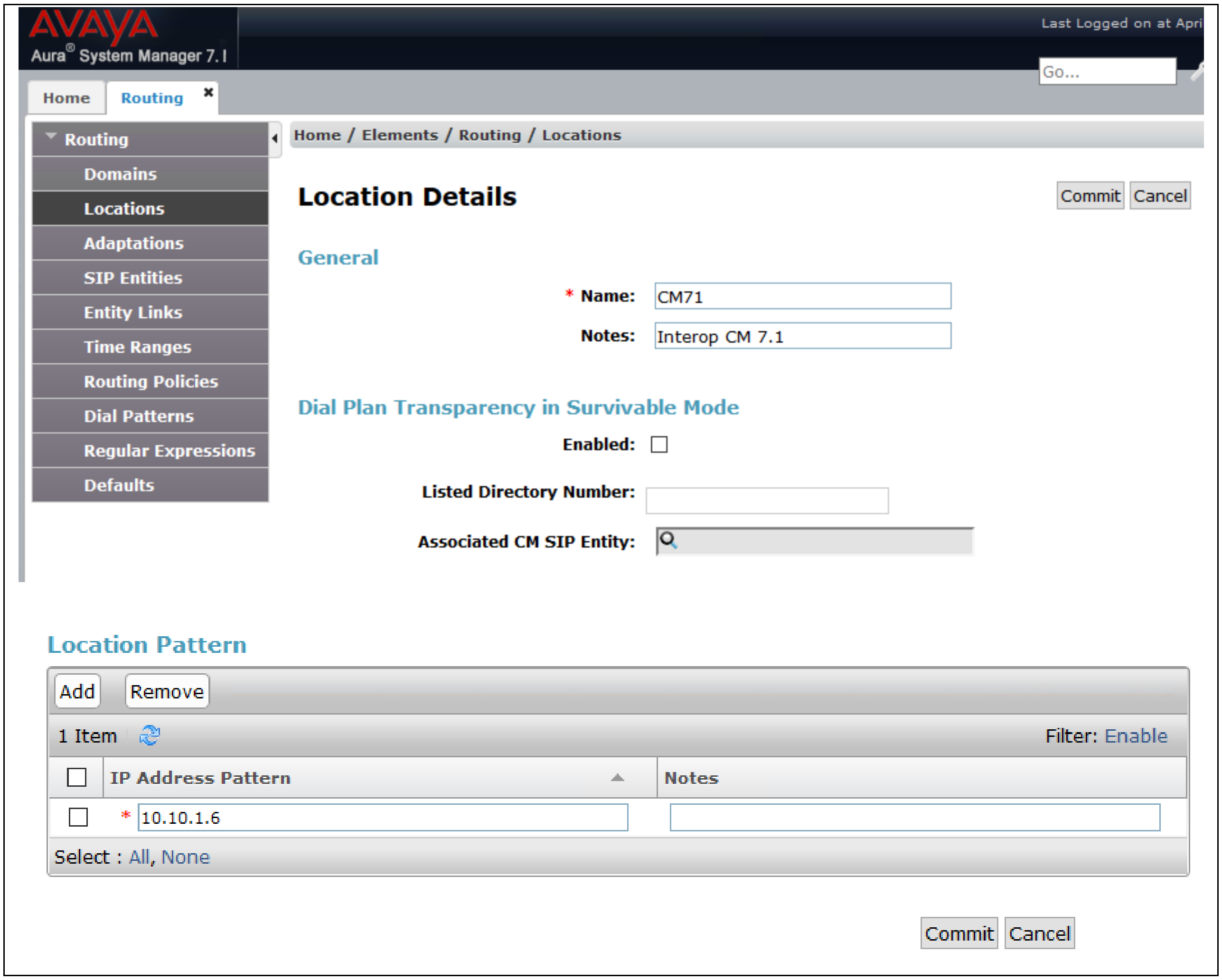


6.2. Configuration of a Location

Click on **Locations** in the left window and if there is no location already configured, then click on **New** to create a new location. However, in the screen below, a location called **CM71** is already setup and click into this to show its contents.

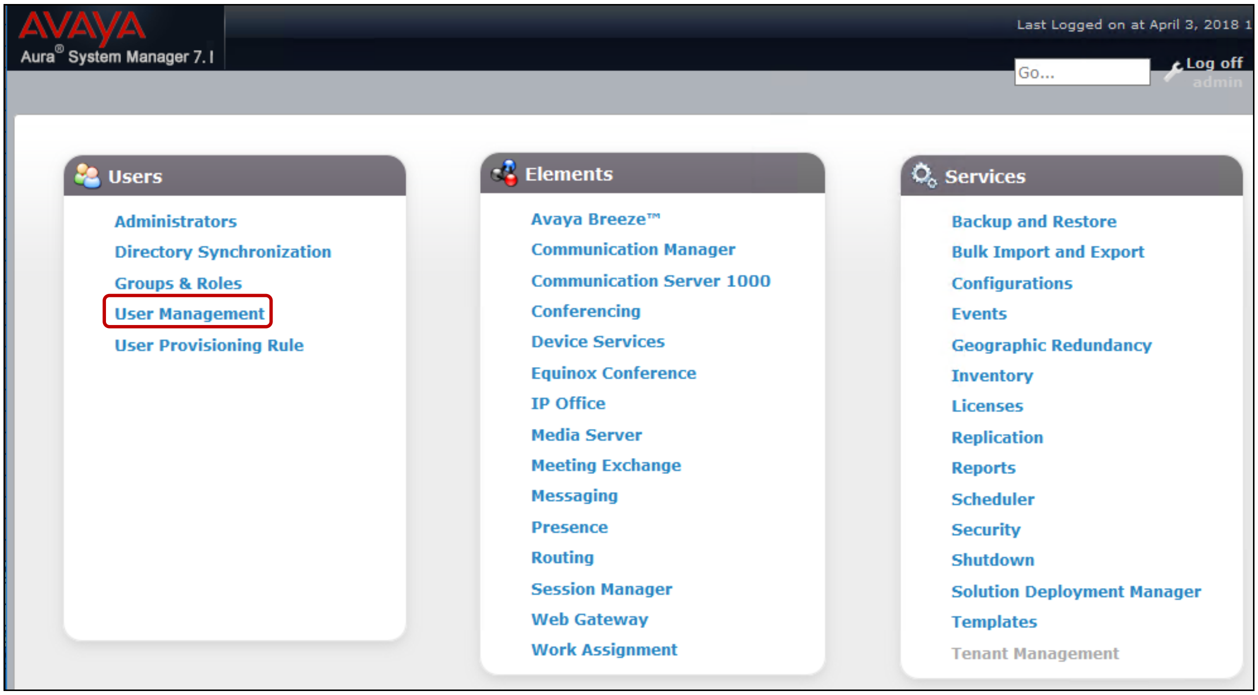


The Location below shows **Name** with **Location Pattern** of **10.10.1.6**. Once this is configured, click on **Commit**.

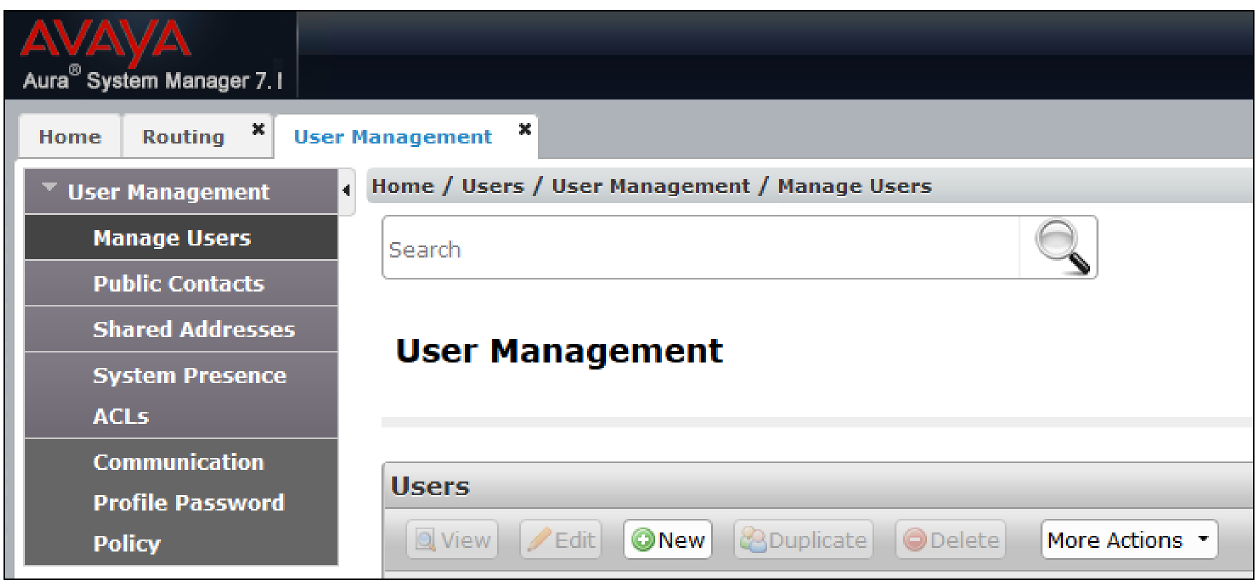


6.3. Adding Ascom SIP Users

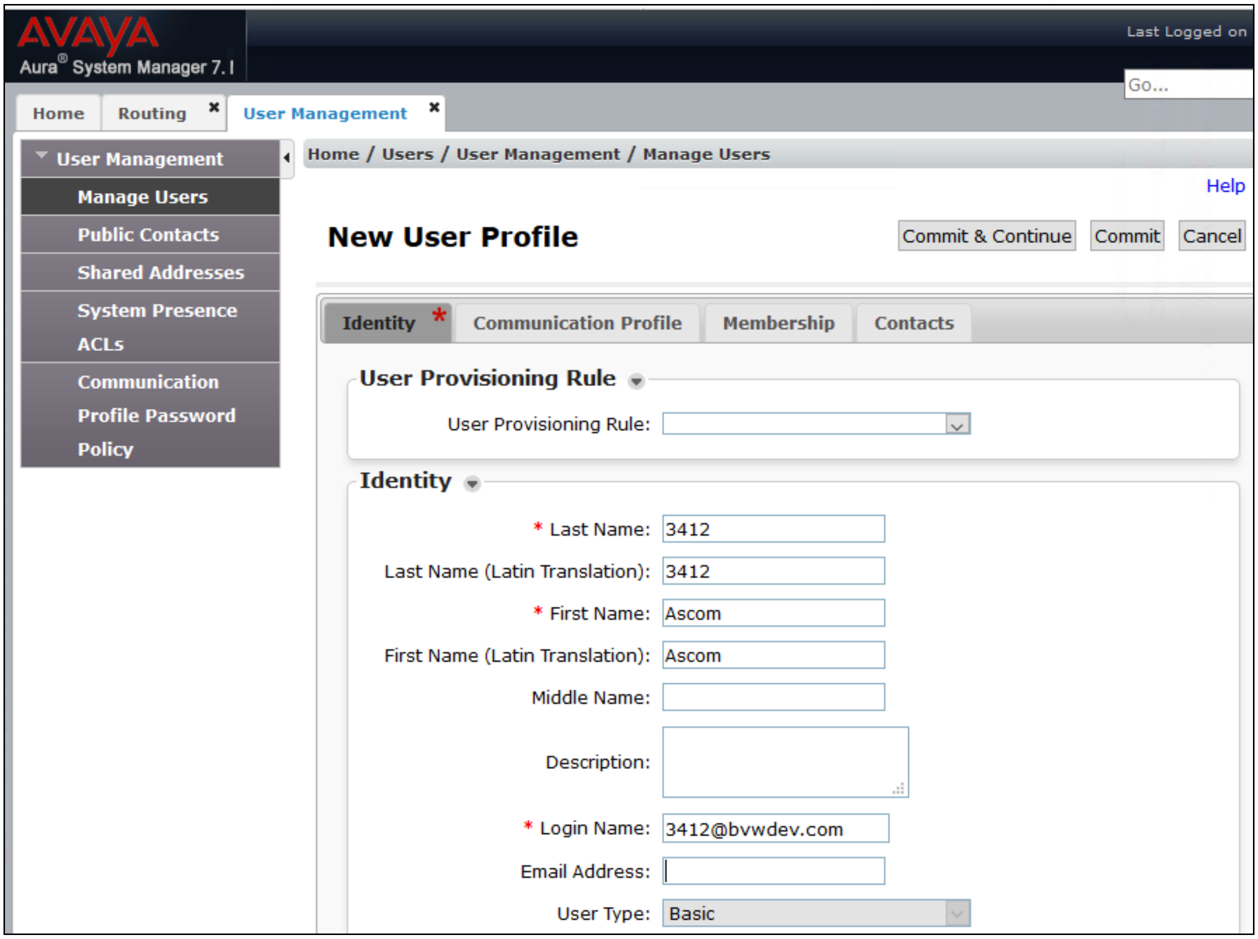
From the home page click on **User Management** highlighted below.



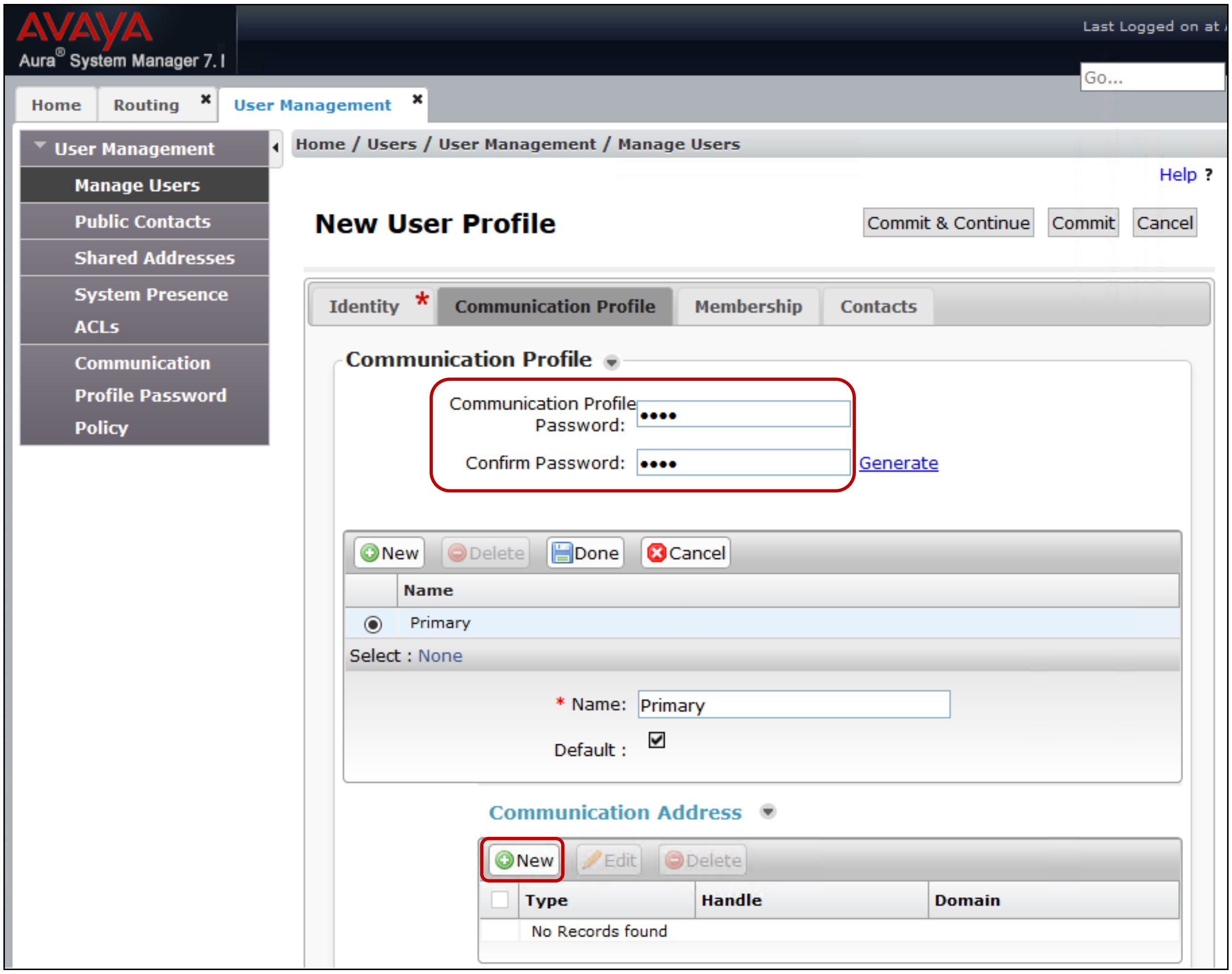
From **User Management** section, click on **New** to add a new SIP user.



Under the **Identity** tab fill in the user's desired **Last Name** and **First Name** as shown below. Enter the **Login Name** following the format of "user id@domain". The remaining fields can be left as default.

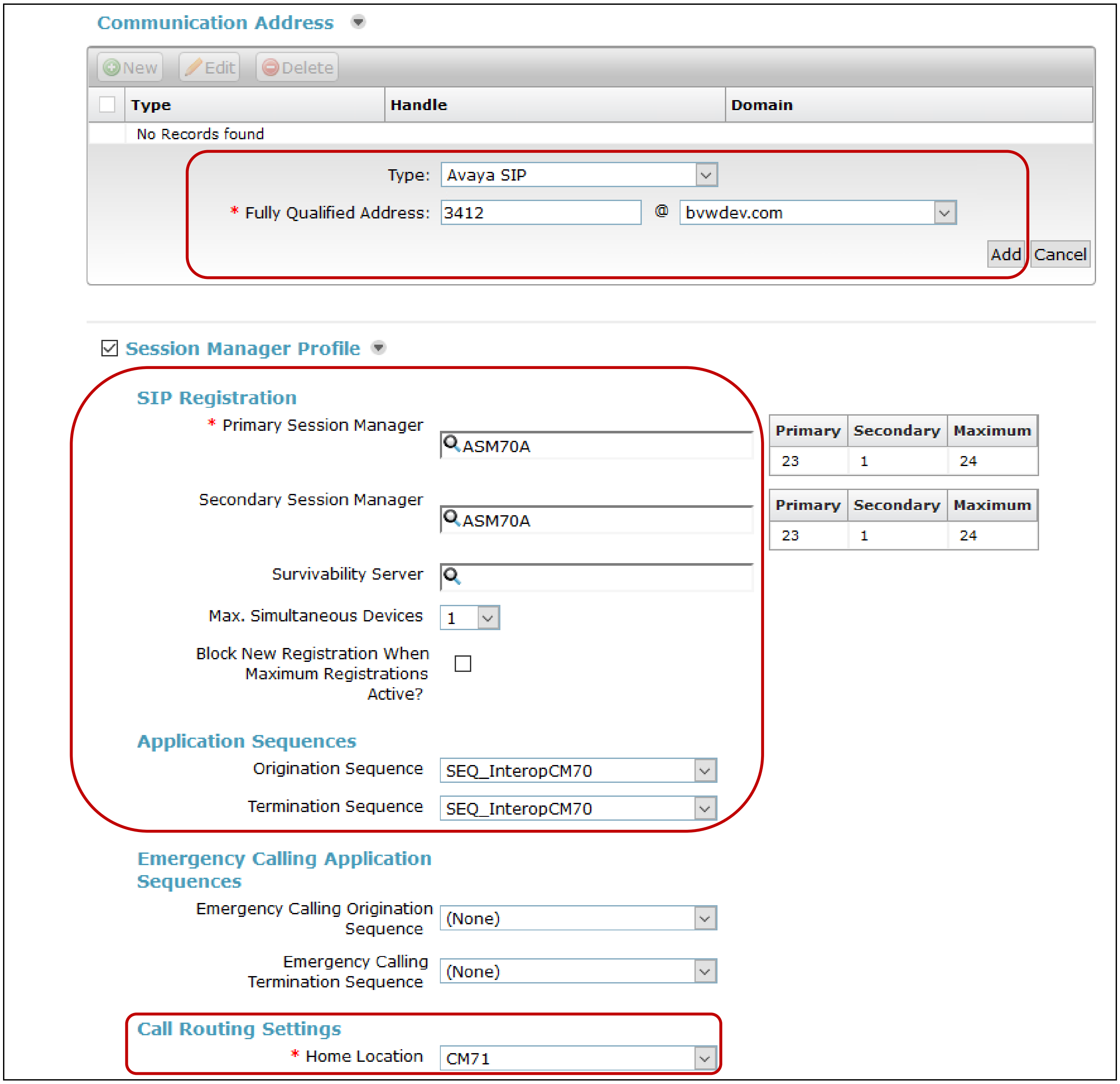


Under the **Communication Profile** tab enter **Communication Profile Password** and **Confirm Password**, note that this password is required when configuring the i62 handset in **Section 7.1**. Click on **New** to add a new **Communication Address**.

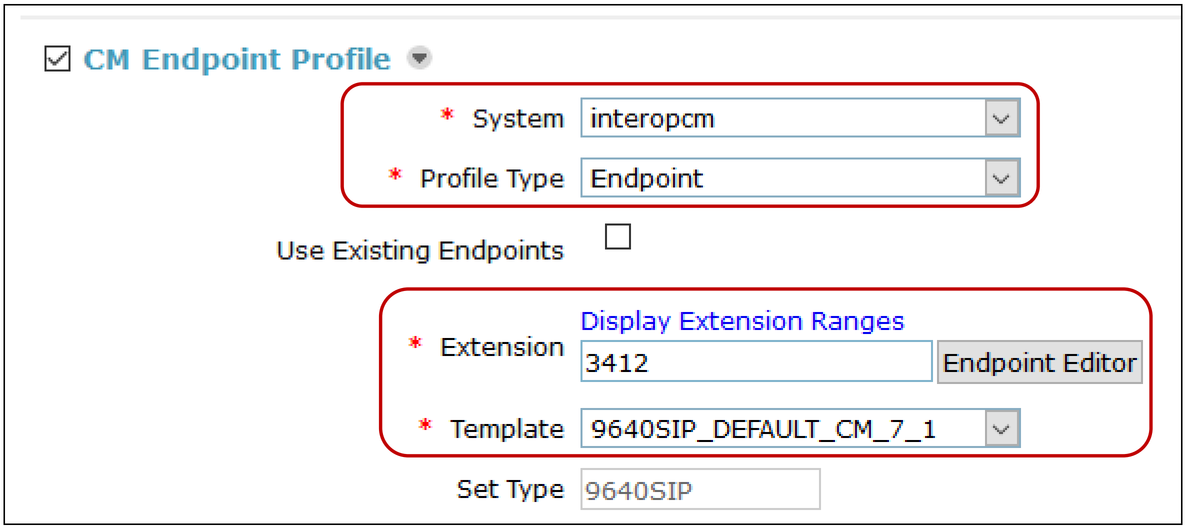


Enter the extension number and the domain for the **Fully Qualified Address** and click on **Add** once finished.

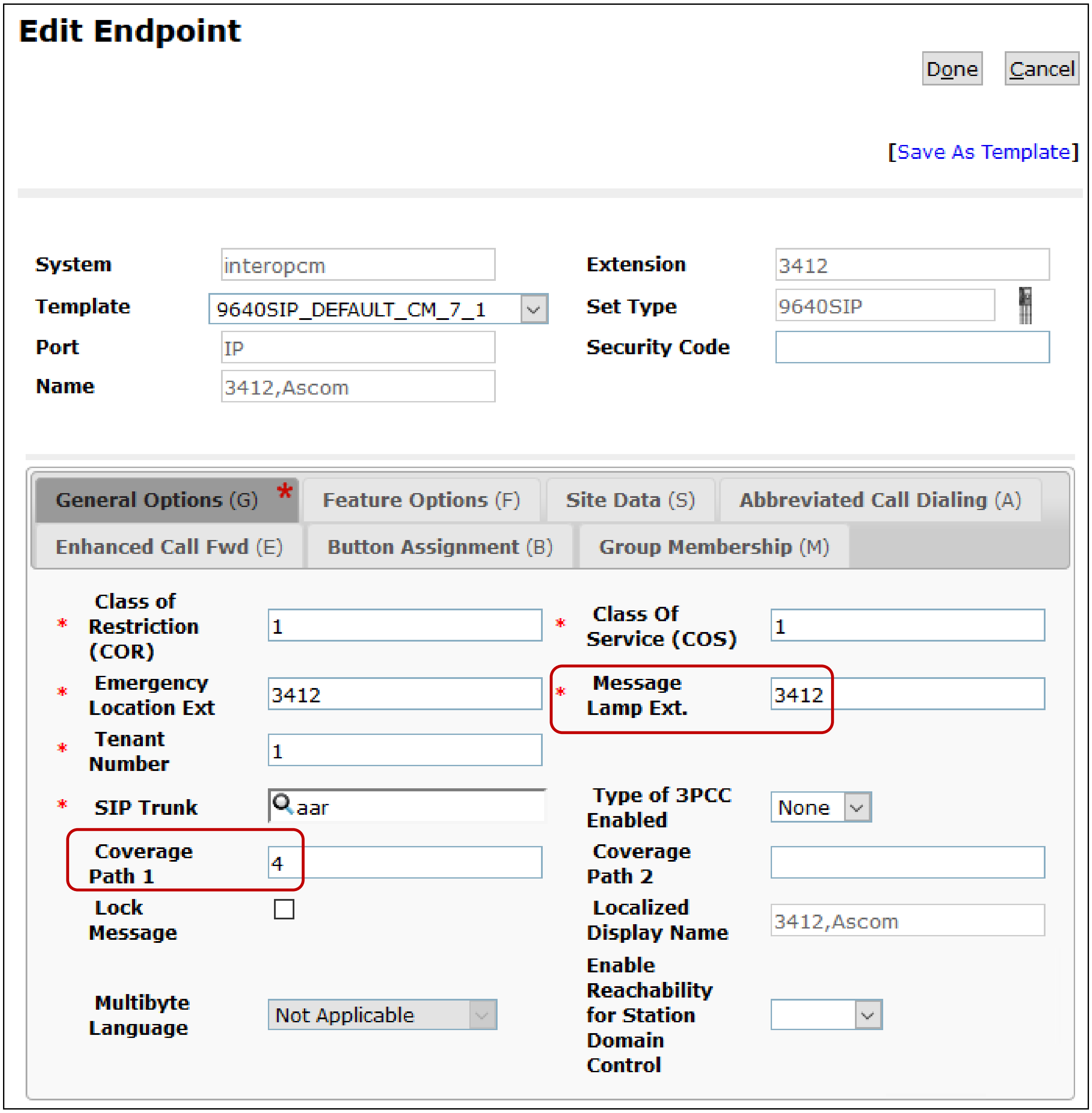
Ensure **Session Manager Profile** is checked. Select pertinent values for **Primary Session Manager**, **Origination Application Sequence**, **Termination Application Sequence** and **Home Location** as highlighted below. Note that if this user needs to support MDA feature, then the **Max. Simultaneous Devices** value should be set as required.



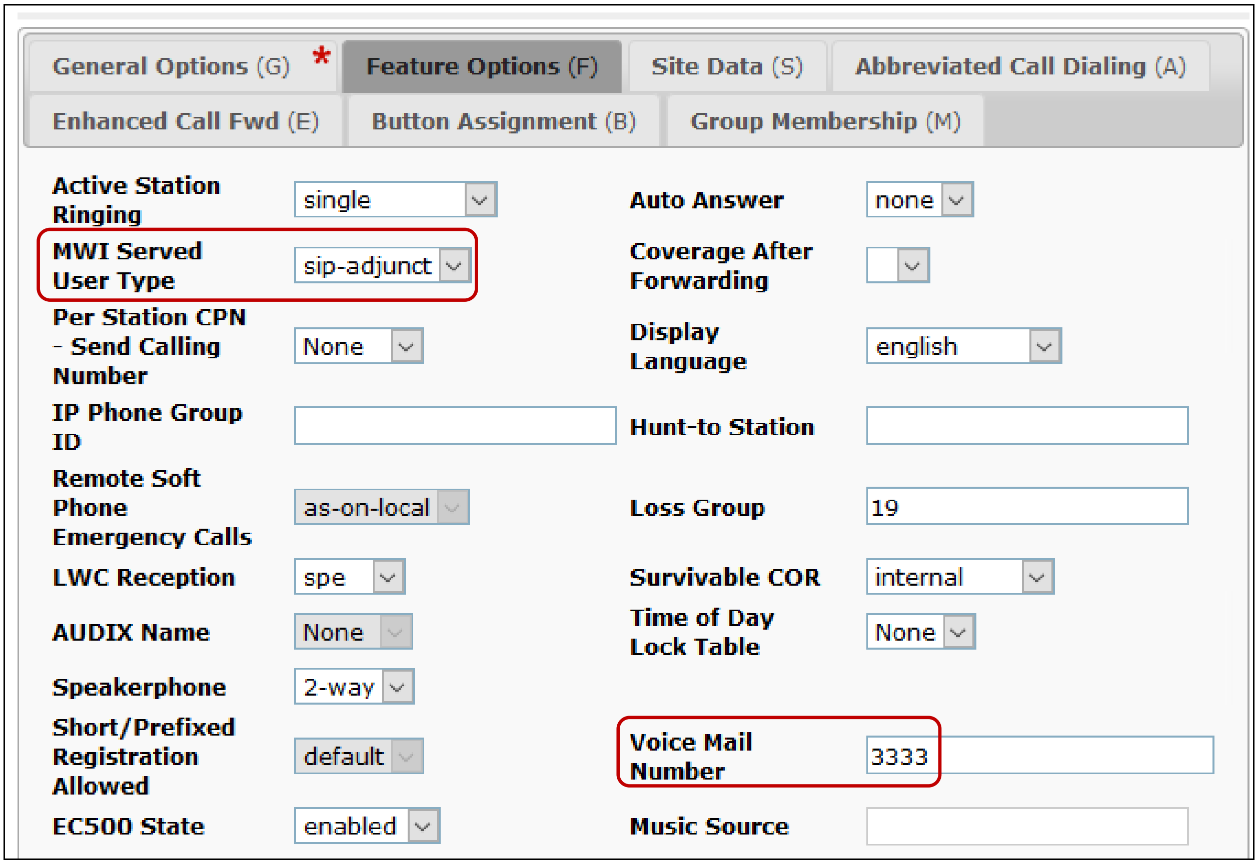
Ensure that **CM Endpoint Profile** is selected for the **System** and choose the **9640SIP_DEFAULT_CM_7_1** as the **Template**. Click on **Endpoint Editor** to configure the buttons and features for that handset.



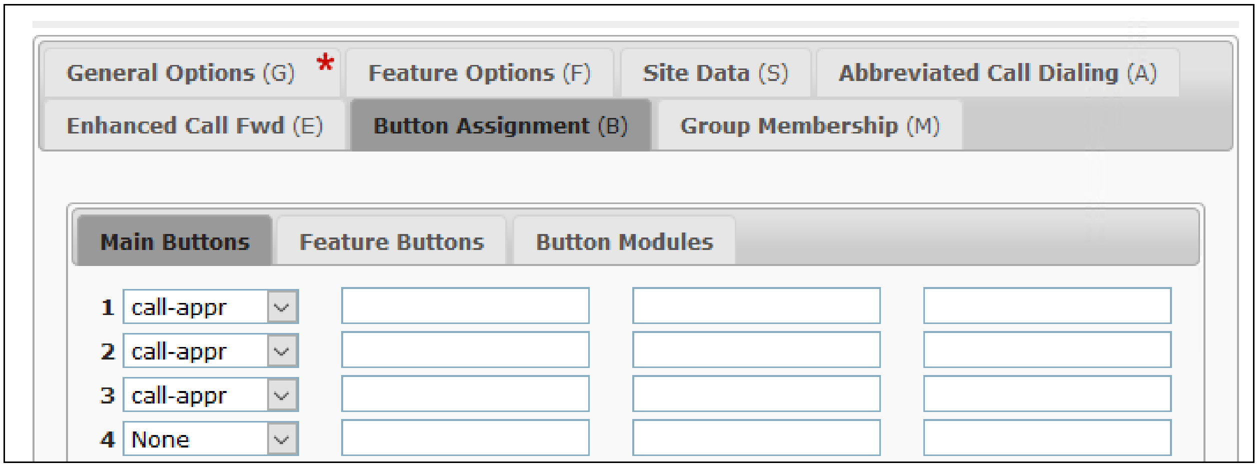
Under the **General Options** tab ensure that **Coverage Path 1** is set to that configured in **Section 5.5**. Also ensure that **Message Lamp Ext** is showing the correct extension number.



Under the tab **Feature Options** ensure that **MWI Served User Type** is set to **sip-adjunct**. Ensure the **Voice Mail Number** is set to that configured in **Section 5.5**.



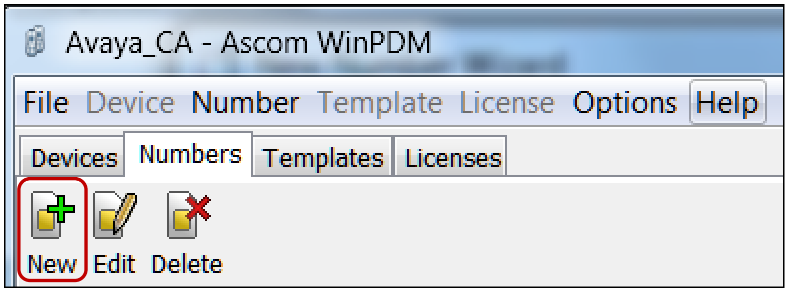
There must be 3 call appearances setup for the i62 handsets for Call Waiting to work.. Once the **Button Assignment** is completed, click on **Done** (not shown) to finish.



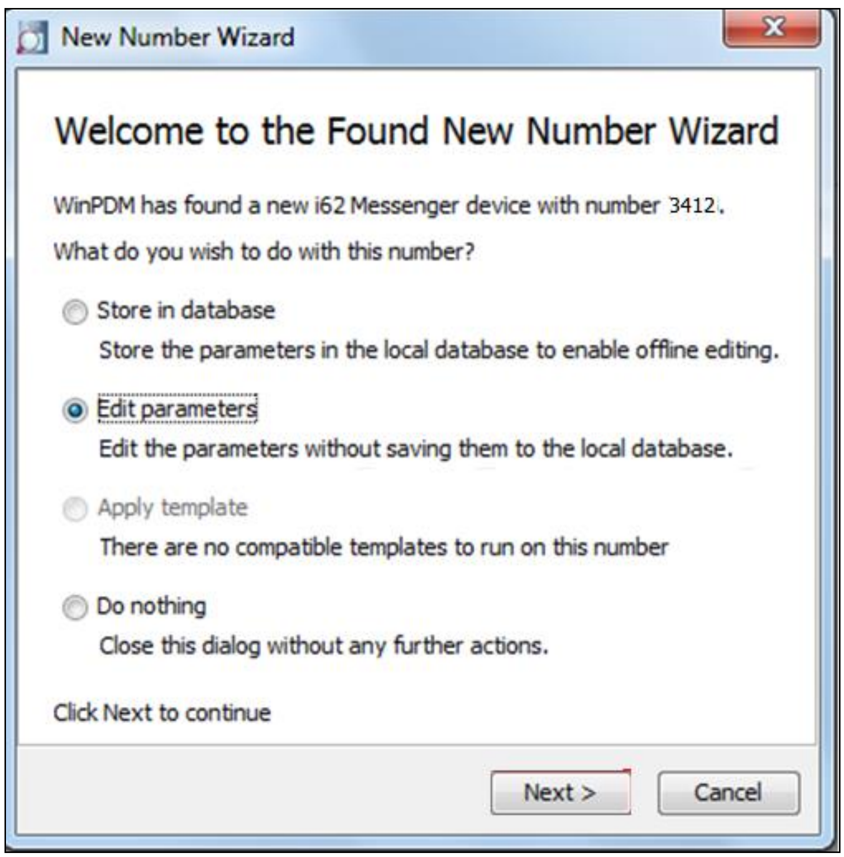
Once the **CM Endpoint Profile** is completed correctly, click on **Commit** (not shown) to save the new user.

7. Configure Ascom i62 VoWiFi Handsets

The configuration of the i62 handsets is done using Ascom's WinPDM software installed on a PC. Attach the Ascom DeskTop Programmer DP1 USB cradle to a PC on which the Ascom WinPDM has been installed. Insert the handset to be configured in the DP1 USB Cradle, start the Ascom Device Manager, select the **Numbers** tab and click **New** icon highlighted below.



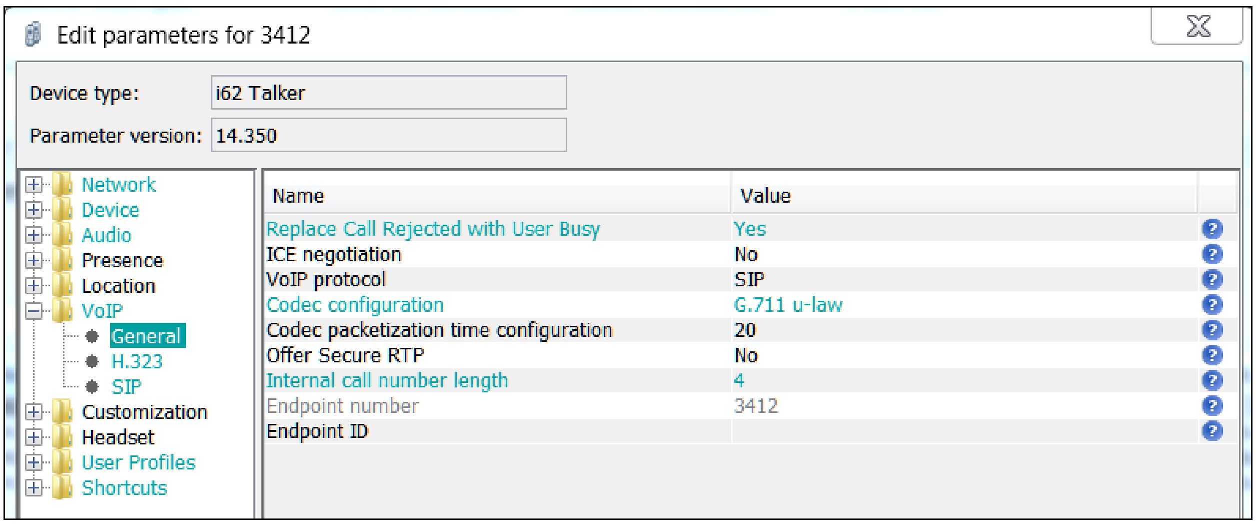
Place a new i62 to be programmed into the cradle and the following screen should appear automatically. Select **Edit parameters** and click on **Next** as shown below.



7.1. Configure SIP settings

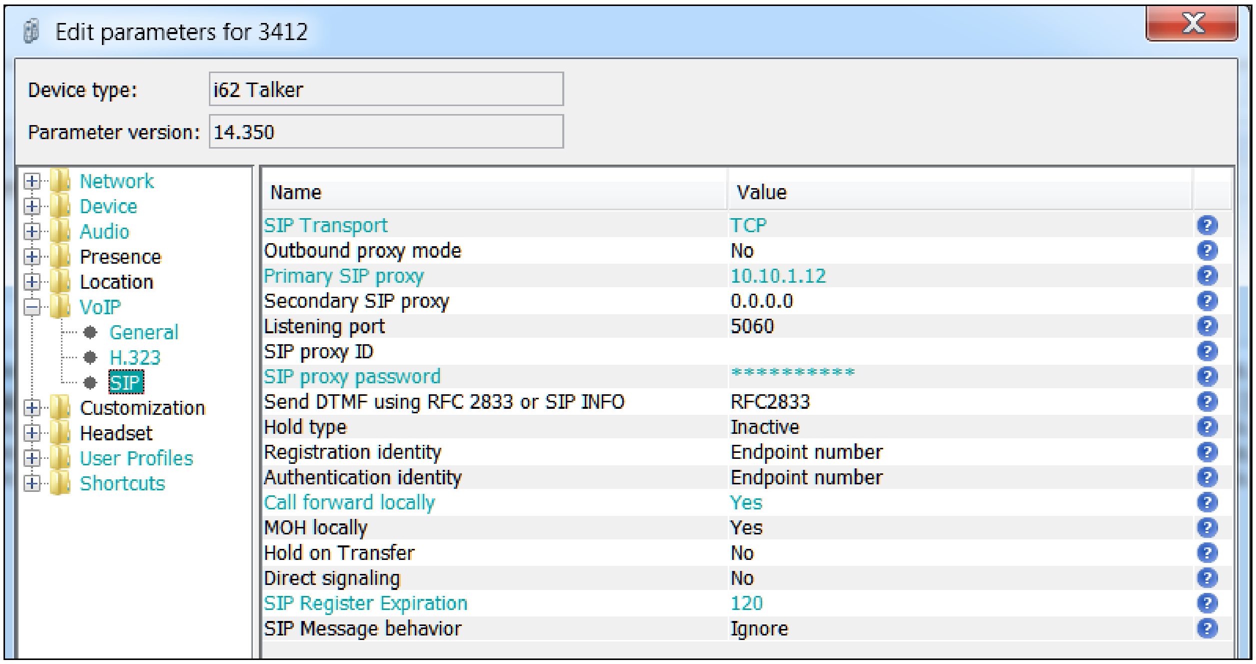
Select **VoIP → General** from the left window. In the main window ensure the following are set.

- **Replace Call Rejected with User Busy:** **Yes**
- **VoIP Protocol:** **SIP**
- **Codec configuration:** **G.711u-law** (as desired based on **Section 5.4**)
- **Codec packetization time configuration:** **20** (as configured in **Section 5.4**)
- **Internal call number length:** **4** (matches #digits in Endpoint number)
- **Endpoint number:** User extension from **Section 6.3**.
- **Endpoint ID:** Can be left blank



Select the **VoIP→SIP** menu point, and enter the values shown below.

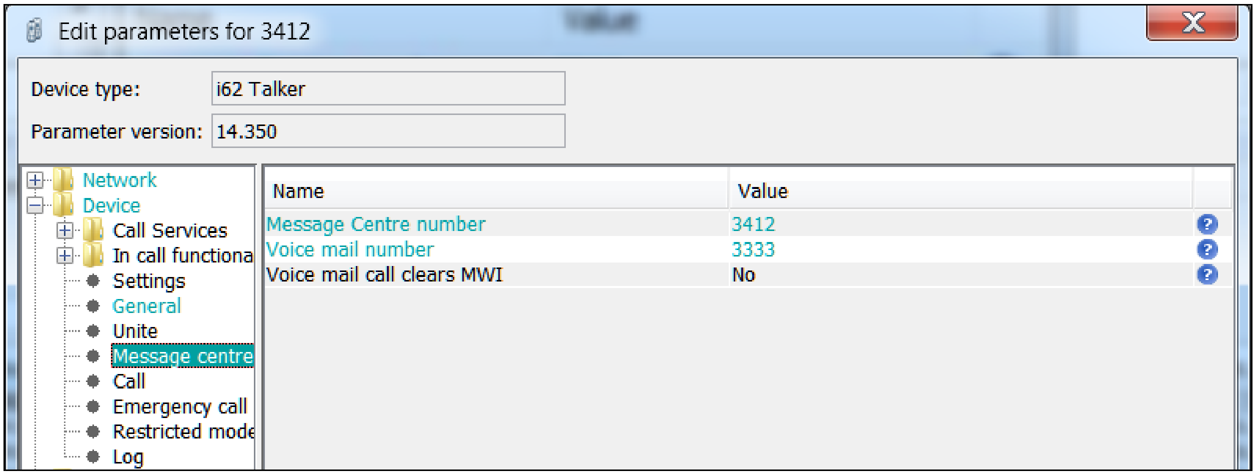
- **Primary SIP proxy:** IP address of Session Manager's signaling interface.
- **Listening port:** **5060**
- **SIP proxy password:** Password assigned to the endpoint in **Section 6.3**.
- **Registration identity:** Enter **Endpoint number**.
- **Authentication identity:** Enter **Endpoint number**
- **SIP Register Expiration:** **120** (recommended value)



For further information about the Ascom i62 handset configurations please refer to Ascom's documentation in **Section 10** of these Application Notes. This section only covers specific settings concerning SIP.

7.2. Configure Message Centre

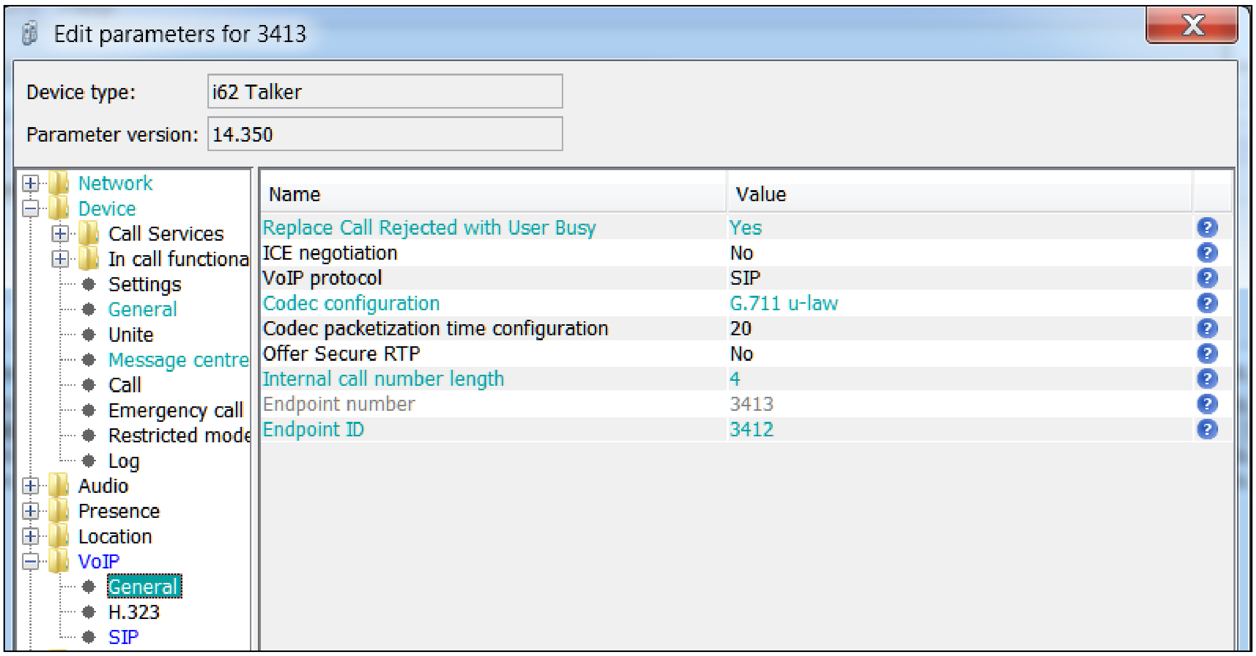
Click on **Device** → **Message centre** in the left window. In the right window, enter the **Voice mail number** as configured in **Section 5.5**and the **Message Centre number** which is the extension number of the handset.



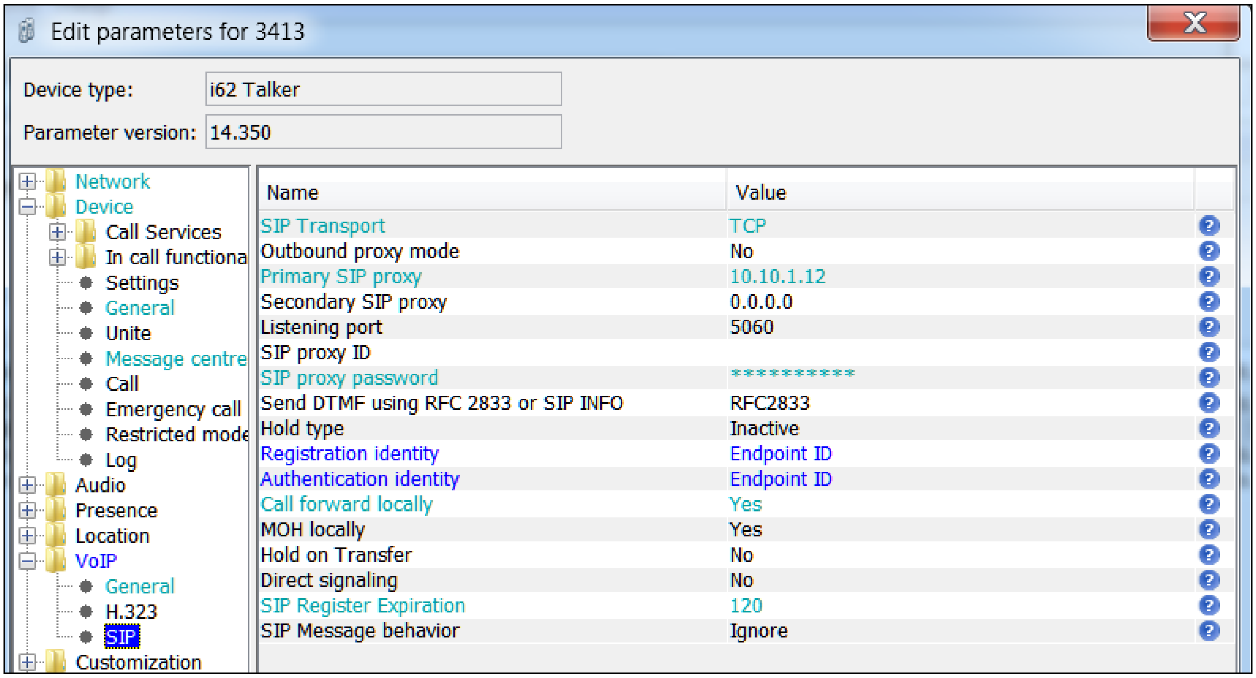
7.3. Configure Multi-Device Access

The MDA feature allows users to leverage multiple devices (endpoints) simultaneously to meet their communication needs. Users can receive and place calls at multiple devices, and move calls between devices as needed.

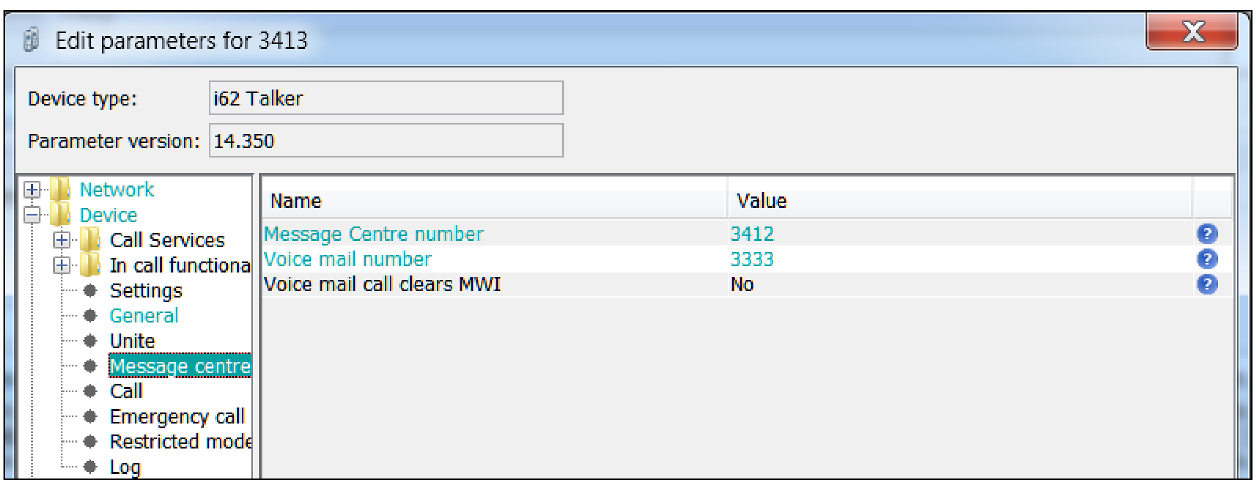
For i62 handset, the MDA feature can be accomplished by configuring and registering the handset using the Endpoint ID parameter. In the example below, handset device with extension number 3413 is configured to register as user 3412. As shown in the screen below, **Endpoint number** is configured as **3413** however **Endpoint ID** is configured as **3412**. As per design, the Endpoint number needs to be unique while configuring the i62 handset via WinPDM.



Also the **Registration identity** and **Authentication identity** are both configured as **Endpoint ID** as shown below.



For the **Message Centre number** instead of the extension number of the handset, configure the Endpoint ID which is **3412** in this case.

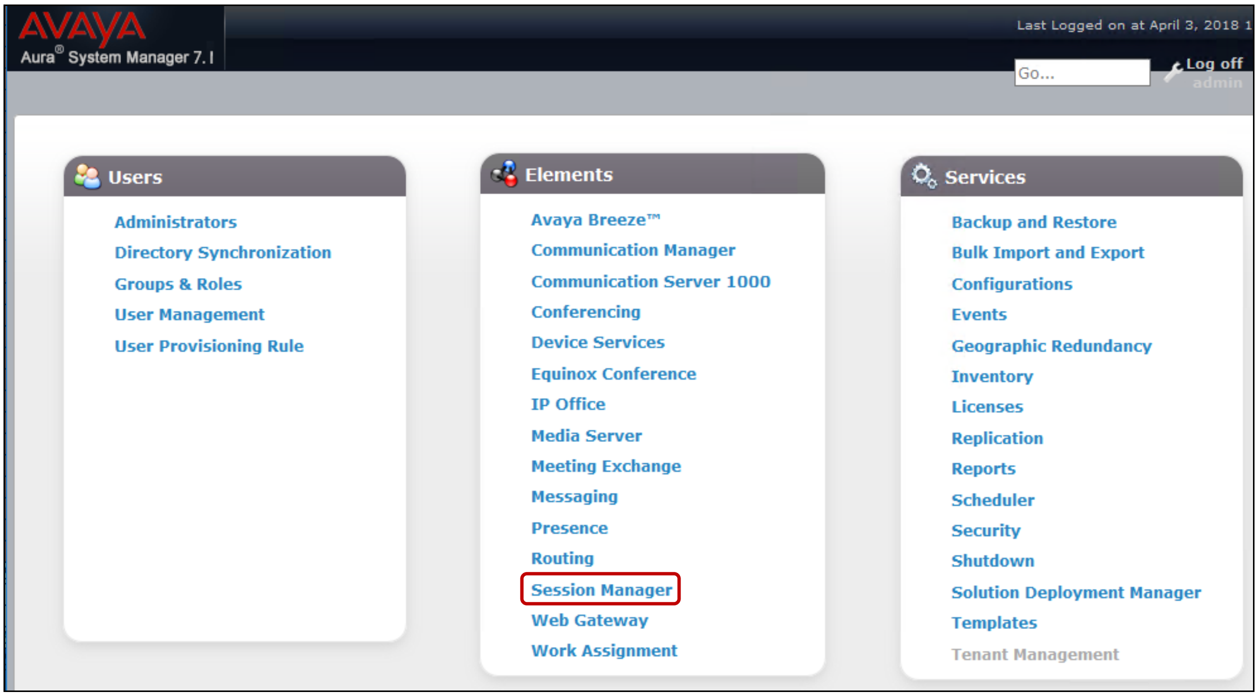


8. Verification Steps

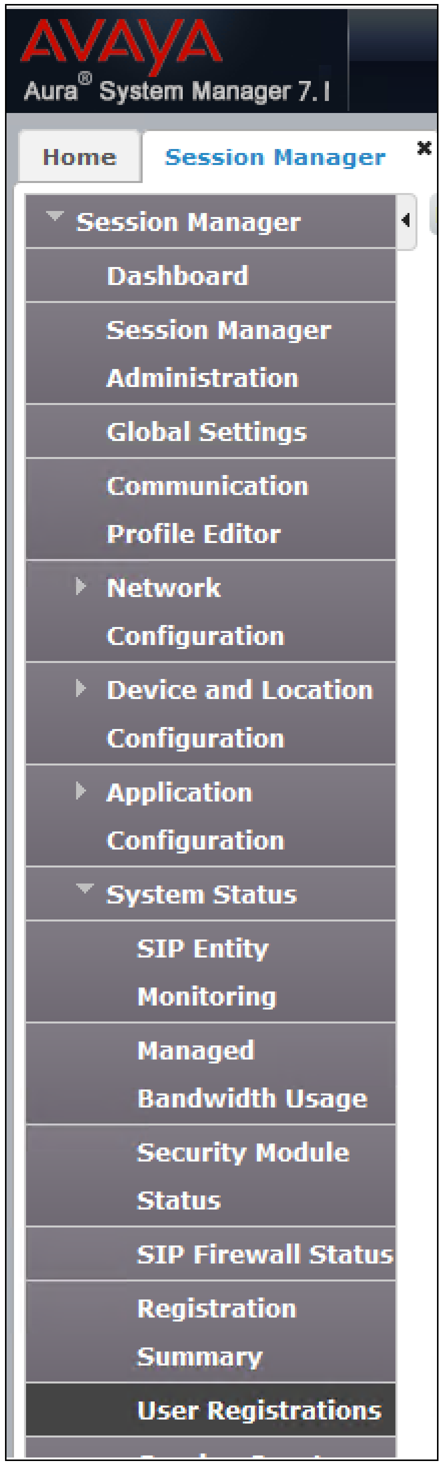
The following steps can be taken to ensure that connections between Ascom i62 handsets and Session Manager and Communication Manager are up.

8.1. Session Manager Registration

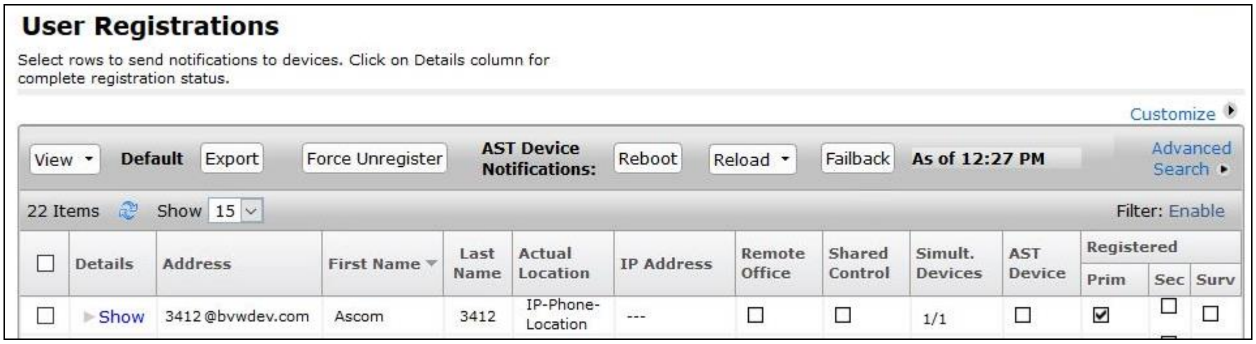
Log into System Manager as done previously in **Section 6**, select **Session Manager** as highlighted below.



Under **System Status** in the left window, select **User Registrations** to display all the SIP users that are currently registered with Session Manager.



The Ascom i62 user **3412** should show as being registered as seen below.



User Registrations

Select rows to send notifications to devices. Click on Details column for complete registration status.

Customize

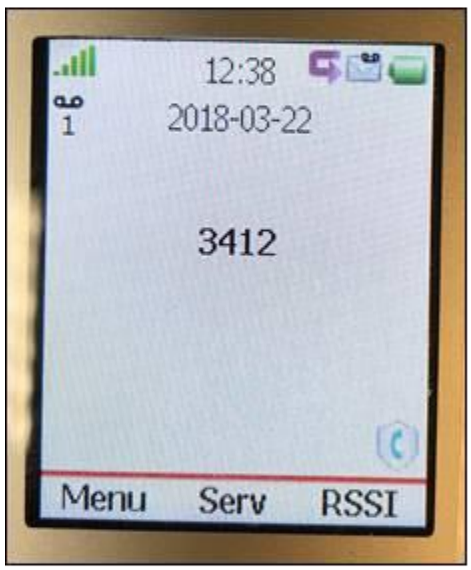
View | Default | Export | Force Unregister | AST Device Notifications: | Reboot | Reload | Failback | As of 12:27 PM | Advanced Search

22 Items | Show 15 | Filter: Enable

	Details	Address	First Name	Last Name	Actual Location	IP Address	Remote Office	Shared Control	Simult. Devices	AST Device	Registered		
											Prim	Sec	Surv
☐	Show	3412 @bvwdev.com	Ascom	3412	IP-Phone-Location	---	☐	☐	1/1	☐	☑	☐	☐

8.2. Ascom i62 Registration

The Ascom i62 handset connection to Session Manager can be verified by an absence of an error message on the handset display just above the red line at the bottom of the display, as shown in the following illustration, (note this is an example from compliance testing).



9. Conclusion

These Application Notes describe the configuration steps required for Ascom's i62 VoWiFi handsets to successfully interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager by registering the Ascom i62 handsets with Avaya Aura® Session Manager as SIP users. Please refer to **Section 2.2** for test results and observations.

10. Additional References

This section references the product documentation relevant to these Application Notes.

Product documentation for Avaya products may be found at http://support.avaya.com.

1. *Deploying Avaya Aura® Communication Manager*, Release 7.1.2, Issue December 2017
2. *Avaya Aura® Communication Manager Feature Description and Implementation*, Release 7.1.2, Issue 4 January 2018
3. *Deploying Avaya Aura® Session Manager*, Release 7.1.2, Issue 4 December 2017
4. *Administering Avaya Aura® Session Manager*, Release 7.1.2, Issue 4 March 2018
5. *Deploying Avaya Aura® System Manager*, Release 7.1.2, Issue 6 March 2018
6. *Administering Avaya Aura® System Manager for Release 7.1.2,* Release 7.1.2, Issue 11 March 2018
7. *Deploying Avaya Aura® Messaging using VMware® in the Virtualized Environment*, Release 7.0.0, Issue 2 January 2017
8. *Administering Avaya Aura® Messaging*, Release 7.0.0, Issue 3 January 2018

Product documentation for Ascom products can be obtained from an Ascom supplier or may be accessed at https://www.ascom-ws.com/AscomPartnerWeb/Templates/WebLogin.aspx (login account for the Ascom Partner Extranet required).

Appendix

Signaling Group

```
display signaling-group 1                                       Page   1 of   2
                                SIGNALING GROUP

 Group Number: 1              Group Type: sip
  IMS Enabled? n        Transport Method: tls
        Q-SIP? n
     IP Video? n                                  Enforce SIPS URI for SRTP? n
  Peer Detection Enabled? n  Peer Server: SM
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
Alert Incoming SIP Crisis Calls? n
   Near-end Node Name: procr                Far-end Node Name: interopASM
 Near-end Listen Port: 5061               Far-end Listen Port: 5061
                                       Far-end Network Region: 1

Far-end Domain: bvwdev.com
                                            Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                 RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload           Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                    IP Audio Hairpinning? n
         Enable Layer 3 Test? y                  Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n           Alternate Route Timer(sec): 6
```

Trunk Group

```
display trunk-group 1                                           Page   1 of  22
                                TRUNK GROUP

Group Number: 1                   Group Type: sip           CDR Reports: y
  Group Name: Private Trunk              COR: 1        TN: 1        TAC: #01
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                Night Service:
Queue Length: 0
Service Type: tie                  Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 1
                                                     Number of Members: 14
```

```
display trunk-group 1                                           Page   2 of  22
      Group Type: sip

TRUNK PARAMETERS

     Unicode Name: auto

                                            Redirect On OPTIM Failure: 5000

            SCCAN? n                                 Digital Loss Group: 18
                      Preferred Minimum Session Refresh Interval(sec): 90

 Disconnect Supervision - In? y  Out? y


             XOIP Treatment: auto    Delay Call Setup When Accessed Via IGAR? n




 Caller ID for Service Link Call to H.323 1xC: station-extension
```

```
display trunk-group 1                                           Page   3 of  22
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                         Maintenance Tests? y




   Suppress # Outpulsing? n  Numbering Format: private
                                                UUI Treatment: shared
                                              Maximum Size of UUI Contents: 128
                                                 Replace Restricted Numbers? y
                                                Replace Unavailable Numbers? y

                                                  Hold/Unhold Notifications? y
                                Modify Tandem Calling Number: no
               Send UCID? y



 Show ANSWERED BY on Display? y
```

```
display trunk-group 1                                           Page   5 of  22
                              PROTOCOL VARIATIONS

                                       Mark Users as Phone? n
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n
                       Send Transferring Party Information? n
                                  Network Call Redirection? y
          Build Refer-To URI of REFER From Contact For NCR? n
                                     Send Diversion Header? n
                                   Support Request History? y
                              Telephone Event Payload Type:


                        Convert 180 to 183 for Early Media? n
                  Always Use re-INVITE for Display Updates? n
                        Identity for Calling Party Display: P-Asserted-Identity
            Block Sending Calling Party Location in INVITE? n
                 Accept Redirect to Blank User Destination? n
                                            Enable Q-SIP? n

          Interworking of ISDN Clearing with In-Band Tones: keep-channel-active
                                Request URI Contents: may-have-extra-digits
```

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.